HIGH FREQUENCY PHASE RESPONSE SPECIFICATIONS – USEFUL OR MISLEADING ?

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ABSTRACT

Conventional high frequency phase response specifications of audio electronics, although easily acquired with standard available instruments, are misleading. The **Absolute Phase** value contains two components, **Frequency Independent Delay** and **Frequency Dependent Delay**. It is only the second component, **Frequency Dependent Delay**, which alters the the shape of the audio waveform thus altering the characteristic sound of the processed signal.

To be meaningful, phase data must be presented as two separate values or graphs, **Deviation From Linear Phase** and **Group Delay**. The first is a modified phase graph, where the **Frequency Independent Delay** is subtracted out, leaving only that part of the phase transfer function which alters the shape of the audio waveform. The **Group Delay** graph shows **Delay** vs. **Frequency** for the spectral components of a complex waveform.

This is not a new subject or discovery, but one which needs more of our audio industry's attention. RF, microwave, and acoustic phase measurements have routinely incorporated delay corrections, but conventional phase measurements of audio electronic devices have not corrected for **Frequency Independent Delay** which we propose must be done to make phase specifications meaningful.

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OBJECTIVE OF THIS PAPER

1) To explain why today's conventional phase specifications of audio electronics cannot be used to meaningfully compare devices regarding waveform distortion.

2) To propose revised presentations of the same phase data as two separate parameters, **Deviation From Linear Phase** (over a specified frequency range) and **Group Delay**. The separation of these parameters allows meaningful comparison of the phase transfer functions as they affect waveform distortion.

SCOPE OF THIS PAPER

The scope of this paper is limited to the specification of the phase component of the low pass filter linear gain transfer function of a network. The scope of this paper does not address the high pass filter function, which is considered to be a possible topic for a separate paper.

The term **Phase** here means the phase component of the gain transfer function of a network, that is, a device which has an input and an output.

Phase matching of stereo pairs or multichannel systems is not the issue here, however it will become clear that such matching should also be broken up into two parts, **Deviation From Linear Phase** and **Group Delay**. A single value of 20 kHz phase specification is not sufficient for matching channels.

ALL NETWORKS ARE LOW PASS FILTERS AND HAVE FREQUENCY INDEPENDENT DELAY

Every audio electronic device, which has an input and an output, can be called a **Network**. Every physical **Network** exhibits a **Low Pass Filter** function, so there is always a high frequency rolloff, even if the bandwidth is higher than 20 kHz. All low pass filter functions have a **Frequency Independent Delay** component for the frequency range below the -3 dB corner frequency. This is the **Delay** which must be subtracted out of the **Absolute Phase** data to make it meaningful as a figure of merit for comparison regarding the shape of the audio waveform.

The phase and delay effects of a low pass filter can be significant at frequencies much lower than the magnitude corner frequency. This is one reason why we can hear differences between two otherwise identical circuits of different bandwidths both well above 20 kHz.

CRITERIA FOR DISTORTIONLESS TRANSMISSION

Time Domain – The Physical Form

Sound exists in nature as a **Waveform** which is **Barometric Pressure** vs. **Time**.

A **Perfect Audio Network** exhibits an **Output** waveform which is identical to **Input** waveform.

If we understand that differences in **Gain** and **Time Delay** by themselves do not alter the characteristic timbre of the sound, it is only the **Shape** of the waveform which concerns timbre.

A **High Fidelity Audio Network** exhibits an **Output** waveform which has the identical **Shape** of the **Input** waveform.

Frequency Domain – A Mathematically Related Equivalent

Even though our goal is to achieve an **Output** waveform which has the identical **Shape** of the **Input** waveform, we can use standard **Frequency Domain** measurements of **Magnitude** and **Phase** because they are mathematically equivalent, as explained by Messrs. Fourier and Laplace.

From the **Frequency Domain** point of view, **Distortionless Waveform Transmission** over a defined frequency range of interest, requires that the linear gain transfer function of a network exhibit:

Criterion #1: Flat magnitude response

Criterion #2: Linear phase response.

Criterion #3: The extrapolation of the low pass filter phase function must intersect DC at zero degrees.

CONVENTIONAL PHASE MEASUREMENTS

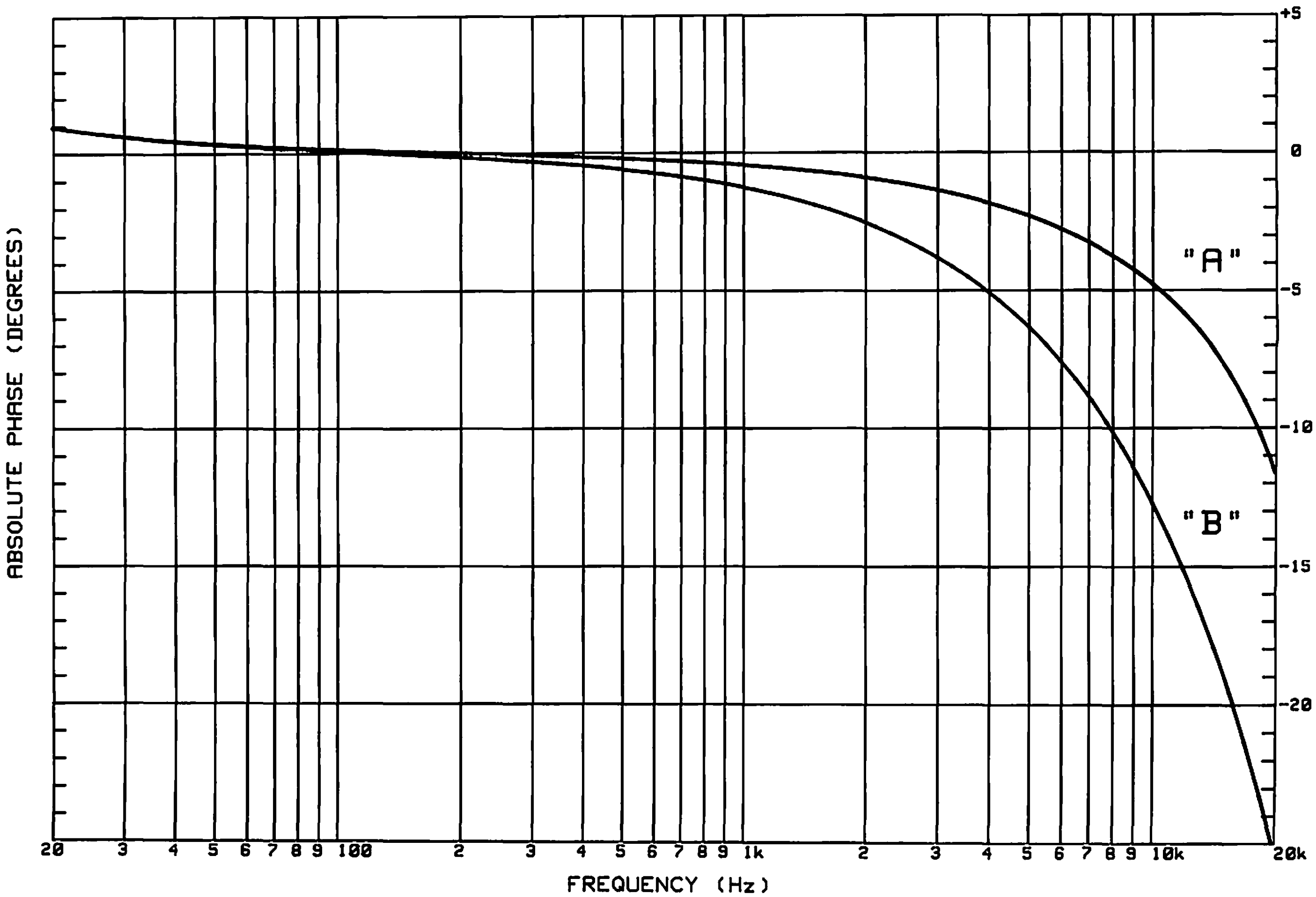
Conventional phase measurements can be duplicated in any laboratory with conventional measurement tools: sine wave generator, oscilloscope, and hand held calculator. We can make such measurements at several frequencies and list a table of numbers or draw a curve on a graph with degrees of phase on the vertical scale and frequency on the horizontal

Absolute Phase measurements are conventionally made by observing the **Absolute** (total) **Time Delay** of a sine wave from the input to the output of the network on a dual trace oscilloscope. Then the number of degrees of **Absolute Phase** is calculated by the direct relationship at that frequency

Absolute Phase = – (**Absolute Time Delay** x **Frequency** x **360**)

where: **Absolute Phase** is in **Degrees** [1] <Equation 1>
Absolute Time Delay is in **Seconds**
Frequency is in **Cycles** per **Second (Hertz)**
the **360** term is **Degrees** per **Cycle** units multiplier

CONVENTIONAL PHASE MEASUREMENTS



"A" PHASE ERROR appears less than "B"

FIGURE 1 – CONVENTIONAL PHASE MEASUREMENTS

Note that network "A" appears to have less phase error than network "B." If the 20 kHz phase specifications are compared, network "A" with 11.5 degrees might be chosen because it shows less **Absolute Phase** than network "B" with over 25 degrees.

But actually, network "B" with the larger **Absolute Phase** number (the more negative value) alters the shape of the audio waveform less. Why?

THE PROBLEM WITH CONVENTIONAL METHODS

The real problem is that the **Absolute Phase** values are derived from **Absolute Time Delay** measurements. This **Absolute Time Delay** is the sum of two delays which need to be considered separately. They are **Frequency Independent Delay** and **Frequency Dependent Delay**.

The **Frequency Independent Delay** is that part where all frequencies are delayed equally. This aspect of the transfer function does not affect the shape of the audio waveform of the processed signal. It simply delays all frequencies equally, as in playing back an ideal recording days after it was recorded. That delay, in itself, does not change the shape of the audio waveform.

The **Frequency Dependent Delay** is that part of the phase measurement where some frequencies are delayed more than others. It is only this delay which must be considered when evaluating how the phase component of the transfer function of a network alters the shape of the audio waveform.

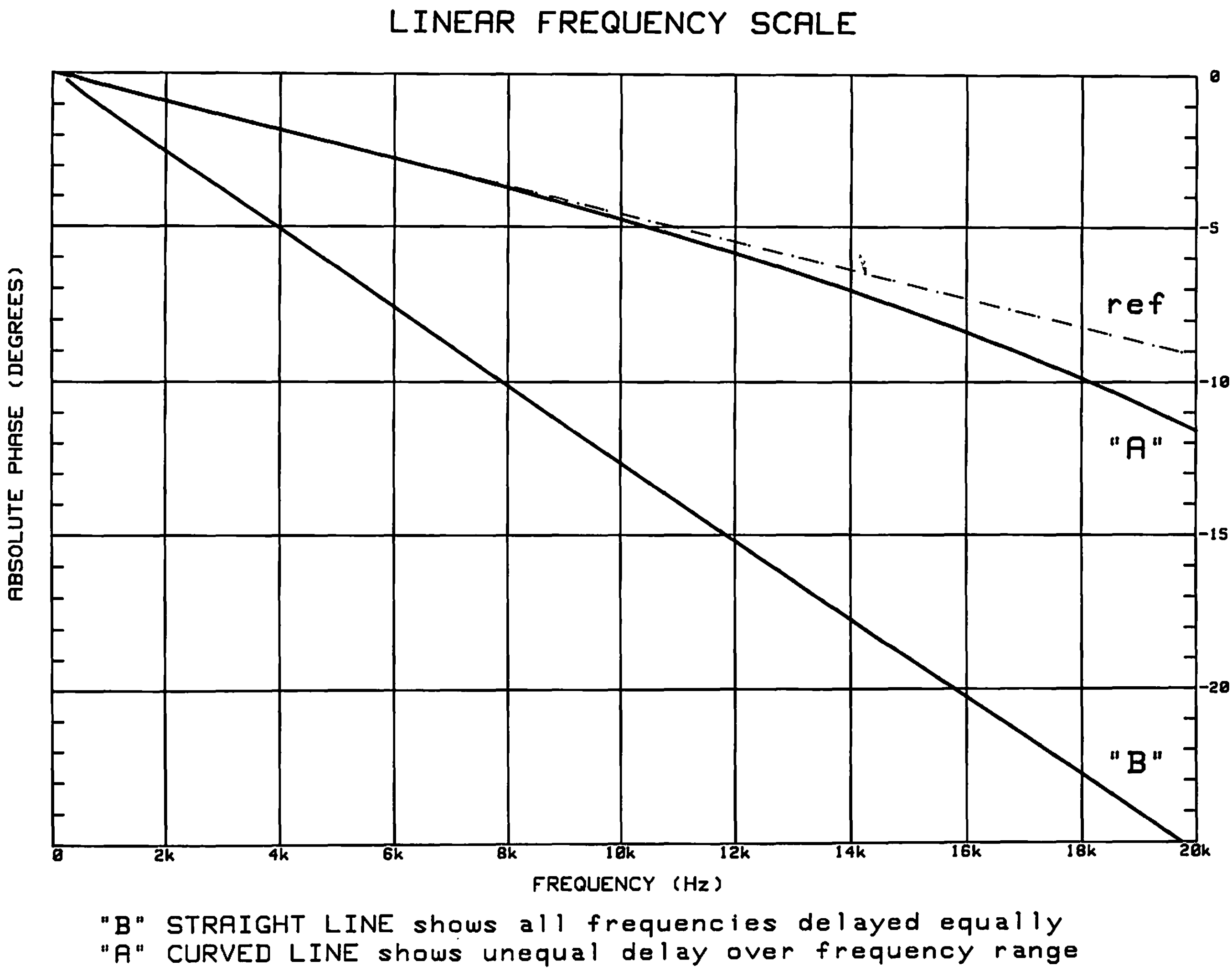


FIGURE 2 - LINEAR FREQUENCY SCALE

Consider this example:

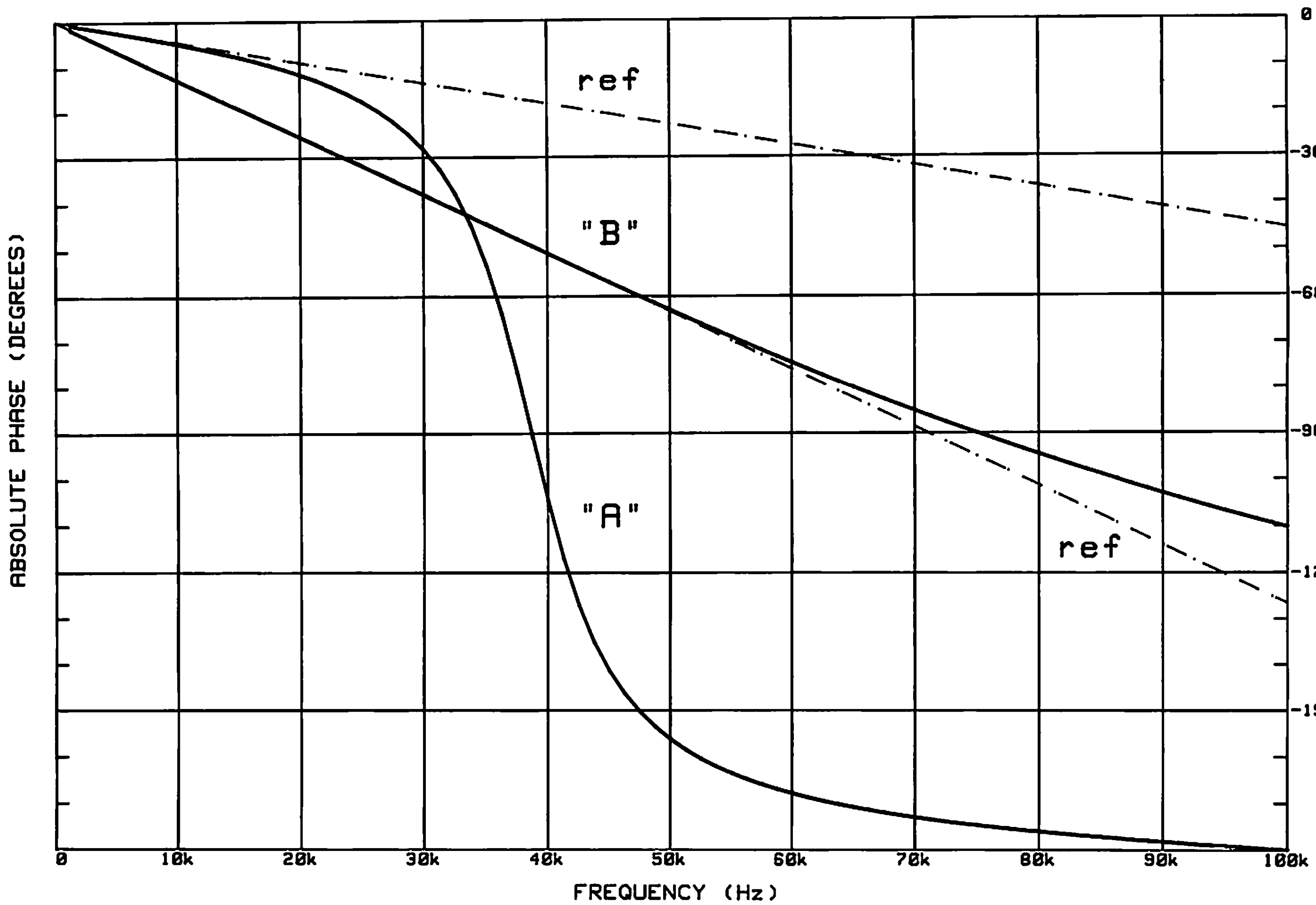
2.778 microsecond **Frequency Independent Delay**, if converted directly to **Absolute Phase** using equation 1, yields 1 degree per kHz which corresponds to this **Linear Phase** relationship:

5 degrees at 5 kHz
10 degrees at 10 kHz
20 degrees at 20 kHz

Note that this **Frequency Independent Delay** with its corresponding **Absolute Phase** values has *no effect* on the **Shape** of the waveform.

Figure 2, the **Linear Frequency Scale** version of the same phase data reveals that the phase response of network "B" is a straight line. This means that there is **Equal Delay** over the frequency range shown. Note how network "A" deviates from the dashed straight line reference. This means that "A" has unequal delay over the frequency range shown. This is our first indication that "B" has actually caused less alteration of the shape of the audio waveform than "A." Note that Figure 1 was misleading.

EXTENDED FREQUENCY RANGE



"B" shows good phase linearity over wide frequency range

FIGURE 3 - EXTENDED FREQUENCY RANGE

The **Extended Frequency Range** version of the same phase data shows how network "B" has considerably better Phase **Linearity** than network "A" over a very wide frequency range.

The dashed straight reference lines indicate the values of **Linear Phase** tangent with the low frequency regions of each low pass **Absolute Phase** function. The value of the corresponding **Frequency Independent Delay** can be calculated by transposing Equation 1, yielding 1.27 micro-seconds for network "A" and 3.52 microseconds for network "B."

This method determines the value of **Frequency Independent Delay** for low pass filter functions which are DC coupled networks, that is, those with no high pass function. For those networks which include high pass as well as low pass functions, the **Group Delay** calculation, detailed later, must be used to determine the value of **Frequency Independent Delay**.

IS LEAST PHASE IDEAL?

No, actually **Linear Phase** is ideal, if we want all spectral components of the complex waveform to arrive at the same time. So any **Phase** graph, as seen on a **Linear Frequency Scale**, which is a **Straight Line** is ideal, independent of the angle of the line, and independent of the value of **Phase** at 20 kHz. Remember that **Criterion #3** must also be satisfied. In this **Linear Frequency Scale** phase graph, **Criterion #3** can be easily seen in one view by verifying that the extrapolation of the low pass filter phase function through DC intersects zero degrees.

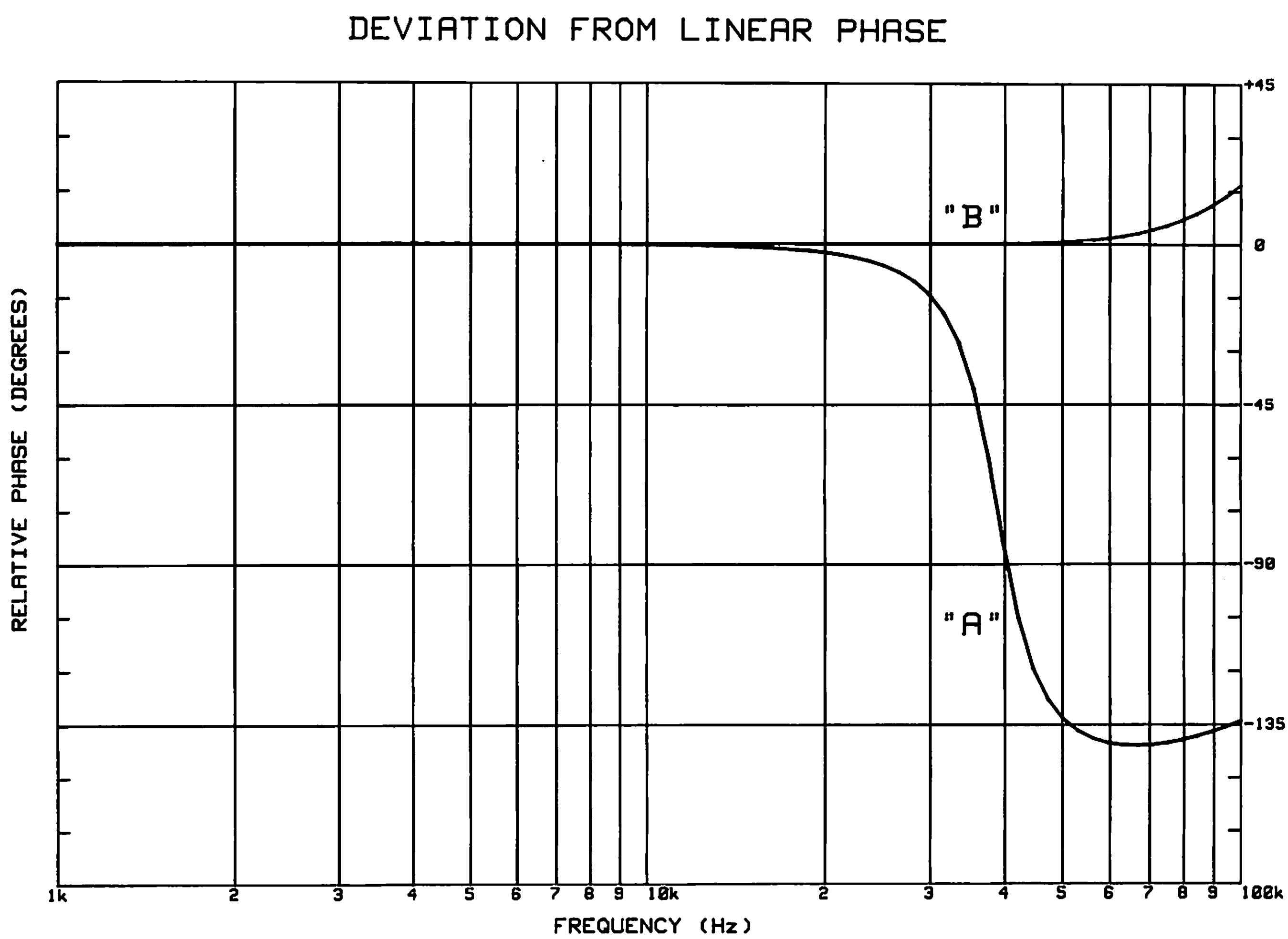
When we are comparing **Absolute Phase** specifications of networks, we have been interpreting the smallest phase value as indicating the least waveform distortion. But the phase value at any frequency by itself is meaningless information, unless we have separated out the **Linear Phase** component. It is the shape of the phase curve that relates to waveform distortion, not the phase value at any point on the curve.

THE SOLUTION: DEVIATION FROM LINEAR PHASE

For **Phase** specifications to be meaningful for comparisons of waveform distortion, the **Frequency Independent Delay** or its corresponding **Linear Phase** component must be determined and subtracted out of the **Absolute Phase** data. This leaves only the **Frequency Dependent** portion of the phase transfer function, which is the only part that affects the waveform shape. We call the resulting specification **Deviation From Linear Phase.**

Deviation = – ((**ABS Time Delay** - **Freq Independent Delay**) x **Freq** x **360**)

where: **Deviation** is in **Degrees** [1] <Equation 2>
ABS Time Delay is in **Seconds**
Freq Independent Delay is in **Seconds**
Freq is in **Cycles** per **Second (Hertz)**
the **360** term is **Degrees** per **Cycle** units multiplier

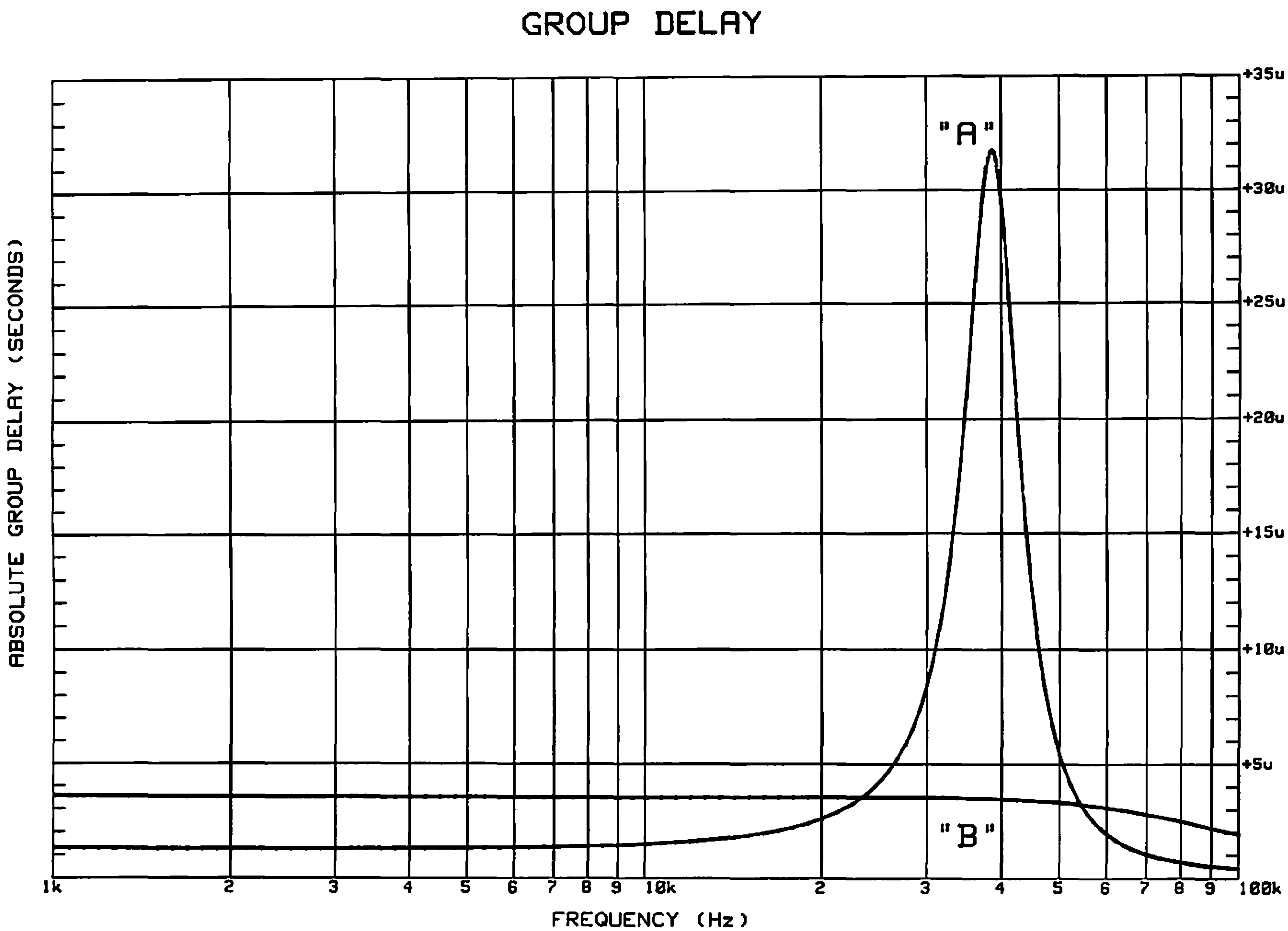


Subtracting the "Frequency Independent Delay" leaves only the "Phase Error" which affects audio waveform

FIGURE 4 – DEVIATION FROM LINEAR PHASE

Now we have a phase graph which represents only that part of the phase component of the transfer function which alters the shape of the audio waveform. It is now very clear that "B" has less **Phase Error** than "A."

So actually, the **Deviation From Linear Phase** graph is the only meaningful representation of the phase component of the low pass filter transfer function which affects the shape of the audio waveform.



Shows delay of spectral components of complex waveform

FIGURE 5 – GROUP DELAY

GROUP DELAY– ANOTHER USEFUL VIEWPOINT

Group Delay is the slope of the **Absolute Phase** function, and it tells us the relative delay of the spectral components of a complex waveform. This is the version of **Delay** vs. **Frequency** which indicates the delay relationships between the fundamental frequency and its harmonics in musical waveforms. When the **Group Delay** is **Flat**, the harmonics arrive at the same time as the fundamental frequency.

To measure **Group Delay** we must measure two (2) values of **Absolute Phase** at two (2) frequencies very close to each other. Then we can calculate **Group Delay**:

> **ABS Phase 1** is the **Absolute Phase** at **Frequency 1**.
>
> **ABS Phase 2** is the **Absolute Phase** at **Frequency 2**.
>
> **Frequency 1** and **2** are equally less and more than the specified frequency by a very small amount

$$\text{Group Delay} = \frac{(\text{ABS Phase 1} - \text{ABS Phase 2})}{(\text{Frequency } 2 - \text{Frequency } 1) \times 360}$$

where: **Group Delay** is in **Seconds** [1] <Equation 3>
ABS Phase is in **Degrees**
Freq is in **Cycles** per **Second (Hertz)**
the **360** term is **Degrees** per **Cycle** units multiplier

In Figure 5, note how flat the **Group Delay** is for network "B", indicating equal delay over the frequency range, and network "A" shows extremely unequal delay. Here you can see that each low pass filter has a flat (frequency independent) delay in the low frequency region. "A" has 1.27 microsecond delay and "B" shows 3.52 microseconds. Although "B" has more delay, it is more **Flat.**

IS FLAT GROUP DELAY IDEAL?

Yes. If the **Group Delay** is flat, all frequencies have been delayed equally. This means that the audio waveform has been shifted in time, but the shape of the waveform has not been altered, thus the characteristic sound of the signal has not been changed.

CAUTION: You must also verify that the **#3 Criterion** for **Distortionless Transmission** is satisfied, i.e. extrapolation of the low pass filter phase function must intersect DC at zero degrees.

BESSEL LOW PASS FILTER FUNCTION – ONE SUGGESTED TARGET

The **Bessel** low pass filter functions exhibit **Linear Phase** and **Flat Group Delay** and therefore are suggested as ideal target functions for audio signal processing. You may have another favorite low pass filter function. The author invites discussion which will focus more attention on the effects of the linear gain transfer function upon sonic clarity.

FIGURES 6, 7, and 8:
MAGNITUDE and STEP RESPONSE

These more conventional methods of evaluating the frequency domain and time domain responses of networks reveal that "A" is a model of a severely underdamped transformer with considerable peaking and ringing, and "B" is actually a transformer model optimized to fit a 2 pole **Bessel** low pass filter function with the typically gradual magnitude rolloff and minimal overshoot. Note that although "B" had the higher **Absolute Phase** value at 20 kHz, it is clear that it caused far less waveform distortion than "A."

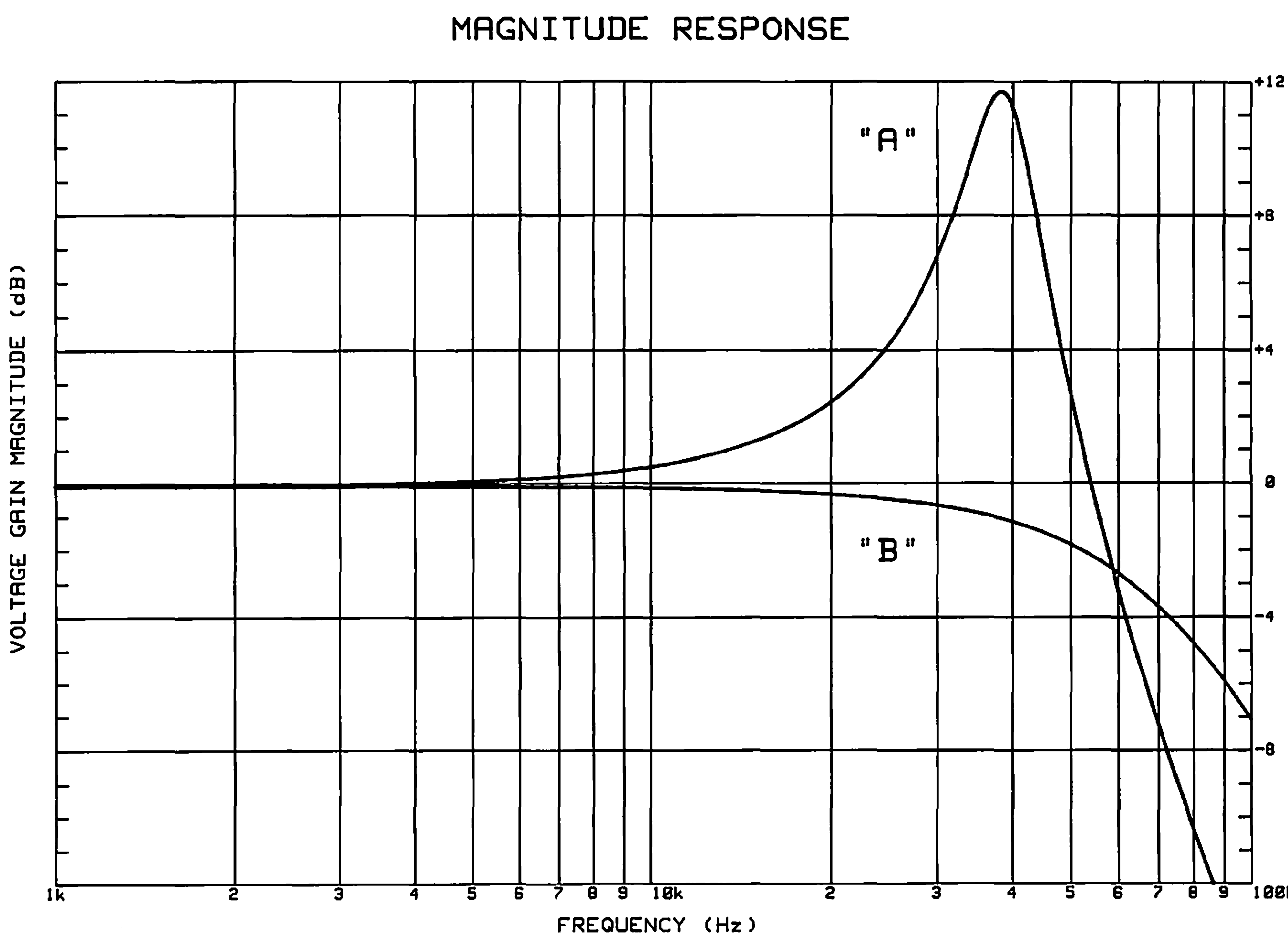


Fig. 6 – "B" shows gentle slope of 2 pole BESSEL low pass filter
"A" shows severe peaking response of underdamped filter

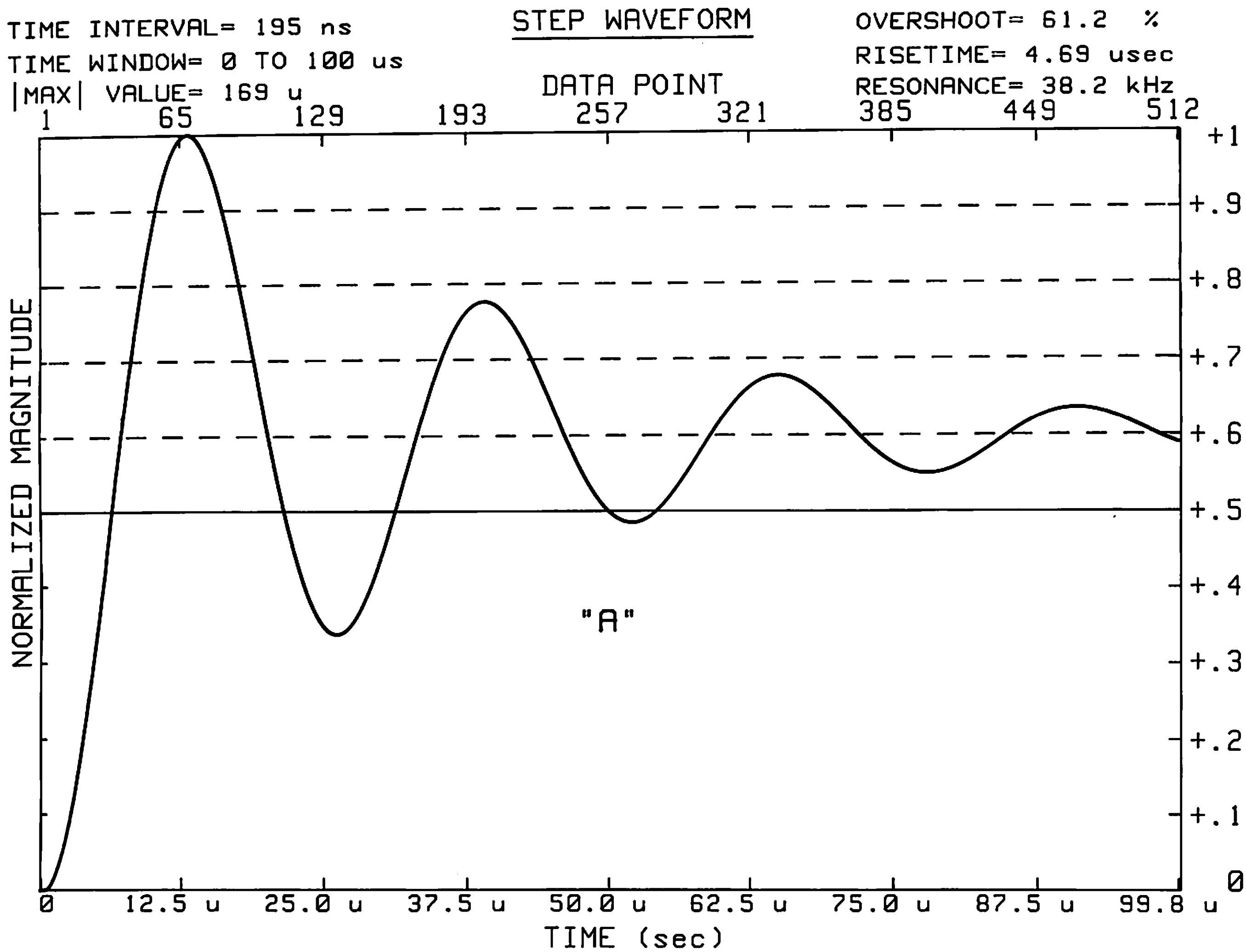


Fig. 7 - "A" shows severe waveform ringing of underdamped filter

FIGURE 7

CONCLUSION

The conventional method of specifying high frequency phase data of audio electronics is meaningless for comparing networks regarding alteration of the shape of the waveform and thus the characteristic sound.

We must adopt the proposed method of specifying **Deviation From Linear Phase** over a specified frequency range and also possibly specify the amount of **Frequency Independent Delay** which was subtracted from the conventional or **Absolute Phase** data to arrive at our result.

Also, it would be useful to show the **Group Delay** graph, as it is another revealing viewpoint.

JENSEN TRANSFORMERS has made a firm commitment to rework all published data sheets for 1987 to show the **Deviation From Linear Phase** and **Group Delay** graphs and numerical specifications so the phase data can be used to evaluate which transformers alter the audio waveform the least.

Until these meaningful presentations of the phase component of the low pass filter functions are available for comparison, the evaluator is advised to cease using conventional phase data as a figure of merit, as it is misleading. Certainly, a numerical phase value at 20kHz is **Meaningless** unless the **Frequency Independent Delay** is subtracted out.

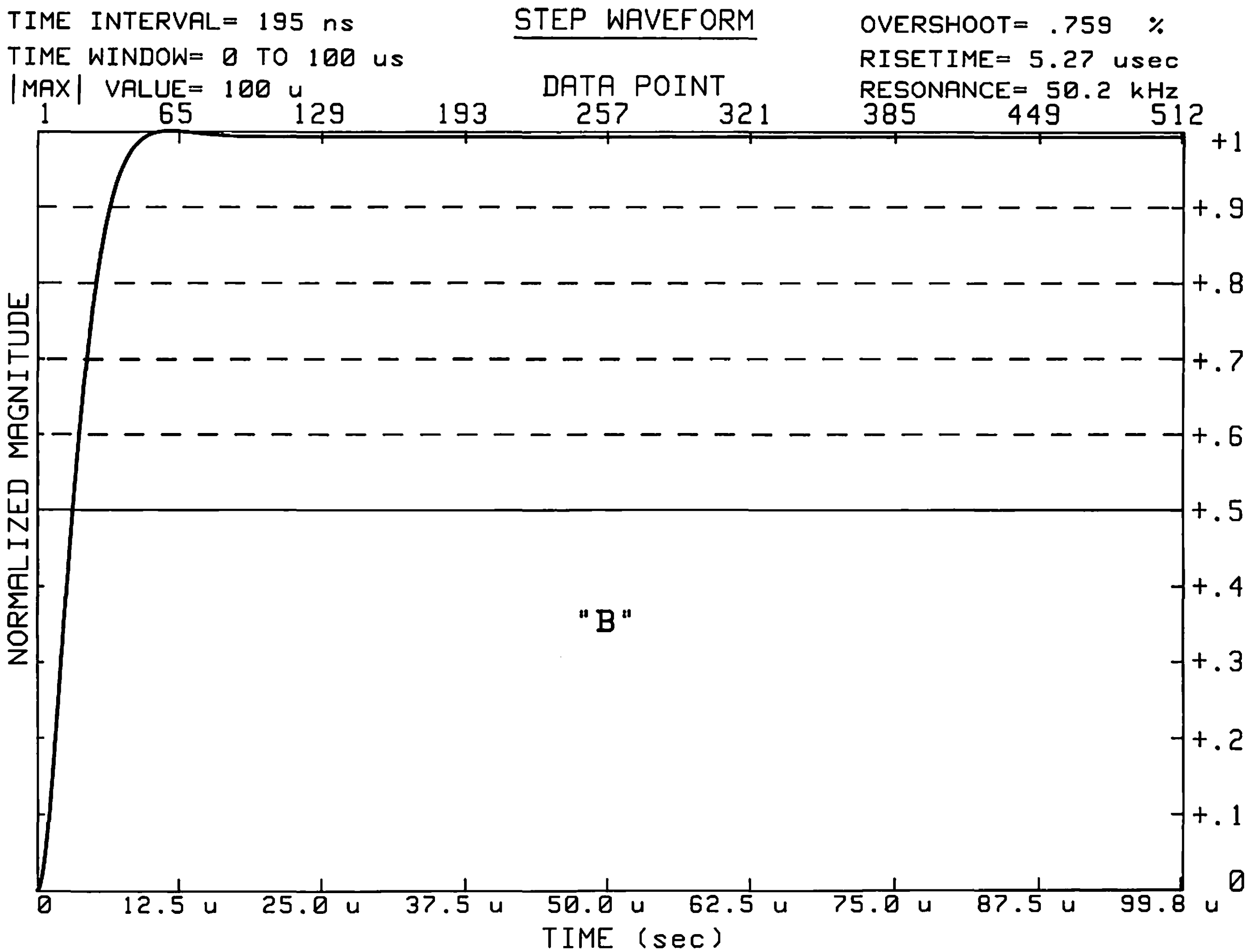


Fig. 8 - "B" shows good waveform response of 2 pole BESSEL filter

FIGURE 8

CONCERNING THE GRAPHS

The graphs were all calculated and plotted directly as the admittance matrix solution of a network using the COMTRAN AC Circuit Analysis program running on a Hewlett Packard 9836 computer and 9872C Graphics Plotter.

REFERENCES

[1] Arthur B. Williams, Electronic Filter Design Handbook (McGraw-Hill, New York, 1981), Chapter 2, pages 2-20 to 2-22.

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ART DESIGN & PRODUCTION

Gary Davis & Associates of Topanga, CA, converted our typewritten information and raw graphs for this technical paper into the printed presentation you see here.